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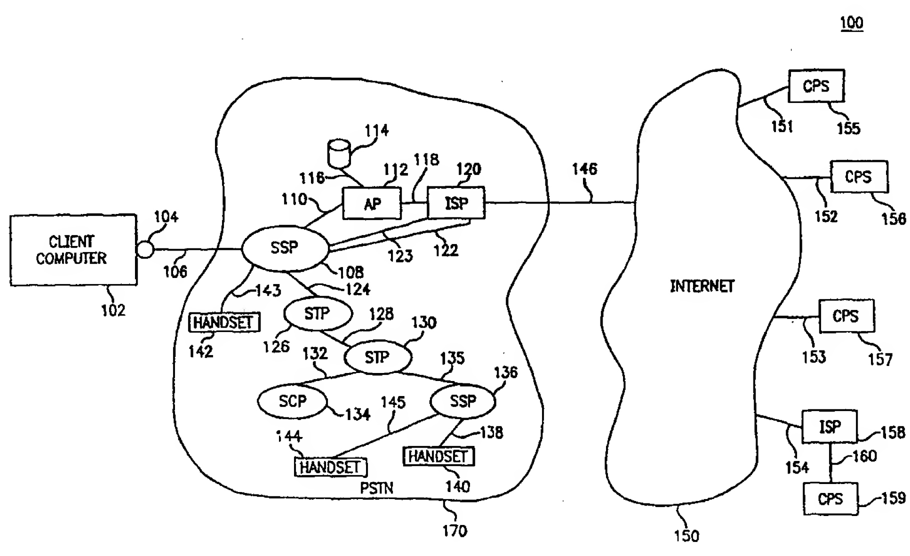

(57) Abstract

A server (112), resident in a public switched telephone network (PSTN) (170) to control communication services associated with a communication switch, includes at least one plain old telephone system (POTS) interface, a data network interface, a storage medium having a plurality of programming instructions stored thereon and an execution unit, coupled to the storage medium, the POTS interface and data network interface, for executing the programming instructions. Wherein the plurality of programming instructions, when executed, implement a plurality of communications services including services to receive the telephone call from the communication switch at the POTS interface and interleavingly deliver incoming call signals to the client computer over the data network, and to accept interleaved outbound call signals from the client computer (102) via data network and to transmit such outbound call signals to an orginating PSTN extension via the POTS interface.

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METHOD AND APPARATUS FOR CONNECTING AN INCOMING CALL TO A COMPUTER SYSTEM

BACKGROUND OF THE INVENTION

1. Related Applications

The present invention is a continuation-in-part of copending Application No. 08/801,698 entitled "Method and Apparatus for Connecting an Incoming Call to a Computer System that is Already Engaged in a Communication Session" filed on February 14, 1997 by George L. Taylor, et al. and commonly assigned to the assignee of the present invention.

2. Field of the Invention

The present invention relates to the field of telecommunications and, in particular, to a method and apparatus for connecting an incoming call to a computer system that is already engaged in a communication session, such as an Internet session.

3. Background Information

Numerous advances have been made in recent years in the field of telecommunications. In particular, the field of internet telephony has been evolving at an increasing rate. Evidence of the evolution of internet telephony is best characterized by the number of products now available in the open market. Products such as Netscape Communicator by Netscape Communications Corporation of Mountain View, California; Internet Connection Phone by International Business Machines of Amonk, New York; Intel Internet Phone (IPhone) by Intel Corporation of Santa Clara, California; NetMeeting by Microsoft Corporation, Redmond, Washington; Quarterdeck WebTalk by Quarterdeck Corporation of Marina Del Rey, California; TeleVox by Voxware Incorporated of Princeton, New Jersey; and WebPhone by Netspeak Corporation of Boca Raton, Florida.

Each of these products offers internet based voice communications with a telephone motif, between two users each using the same (or compatible) product on either end of the internet connection. That is, the internet provides the "switching" architecture for the system, while the computer acts as the "handset", or the audio interface. One reason for the proliferation of these applications is a desire to push the technology of the internet to provide a total communications tool. The appeal to users is that, currently, the use of the internet is free of toll charges. Therefore, a user of an internet phone product may communicate with another user located anywhere else in the world without having to pay the long distance charges associated with making a telephone call using the public switched telephone network (PSTN).

However, consumers expecting to completely eliminate their long-distance telephone bills are in for a disappointment. As they shall soon discover, although innovative in their own right, the current applications identified above have a number of limitations which retard their acceptance as a primary communications tool. One such limitation is that each of the applications identified above requires that both users have installed the same software package, or compatible packages and, therefore, provide a relatively low level of interoperability. One reason for the lack of interoperability between the applications is that the developers of these products have incorporated different voice encoders (commonly referred to as a "voice codec", or simply a "codec") in the products. Accordingly, because of the different codecs used, different applications will generally not recognize speech encoded by a codec of a disimilar application.

Another limitation associated with these products is that they are tied to the internet, often requiring all users to access a common server in order to maintain a directory of available users in which to call. That is to say, the applications identified above do not integrate the packet switched network of the internet with the circuit switched public switched telephone network (PSTN). Therefore, although a computer connected to the internet through an internet service provider (ISP) may communicate with another user on the internet, assuming they are both using a common software application (or at least applications with compatible codecs), these applications do not support communication with a user of a telephone handset.

The reason for this limitation is readily understood by those who appreciate the complexity of the two networks. As alluded to above, the internet is a packet switched network. That is to say, communication over the internet is accomplished by "breaking" the transmitted data into varying-sized packages (or "packets"), based primarily on communication content, and interleaving the various-sized packages to best utilize the bandwidth available at any given time on the internet. When the packets reach their intended destination, they must be reassembled into the originally transmitted data. Loss of packets, and thus data, occur frequently in such a network, and the ability of the network to succesfully transmit information from one point in the network to another determines the quality of the network. For inter-computer communication transactions involving data, the ability to transmit packets and retransmit any packets that are perceived to have been dropped is not a severe limitation and may not even be perceived by the user of the system. However, in a voice communication transaction, the delay required to retransmit even one data packet may be perceived by a user. At best, such delays are an annoying inconvenience.

In contrast to the packet switched network of the internet, the public switched telephone network (PSTN) is a circuit switched network. That is to say that the PSTN assigns a dedicated communication line to a user with which to complete the telephone call, wherein the user can utilize the assigned resource of the PSTN in any way they choose, with the understanding that the user is paying for the use of the dedicated resource of the PSTN. While the circuit switched approach of the PSTN system is not necessarily the most efficient system in terms of call traffic (i.e., it does not make use of the "dead space" common in a conversation), it is relatively easy to ensure that information destined for a particular user is delivered, it simply provides a dedicated line to complete the transaction.

Nonetheless, despite these engineering challanges, a few products are now offered which purport to integrate the PSTN to the internet. Products such as Net2Phone by IDT Corporation of Hackensack, New Jersey, claim to provide a computer user with the ability to place and receive a phone call to/from a PSTN extension. Unfortunately, none of these products completely solve the problem of offering seamless integration of the two networks.

The lack of seamless integration is a limitation keeping prior art internet telephony applications from becoming viable communication tools. An example of such a limitation is associated with the initiation of a call from a remote PSTN extension to a computer system user engaged on the internet. The applications identified above require that, in order to make a phone call from remote PSTN extension to a computer system user via the internet, the user of the remote PSTN extension will have to dial a local internet phone access number, wherein the user will then be asked to dial the internet protocol (IP) address associated with the computer running the internet telephony software to which they are attempting to access.

Although providing an easy technological solution (for the software developers of the application), requiring the typical consumer to remember and dial an IP address is impractical for a number of reasons. First, home based computer users, relying on a dial-in internet service provider (ISP), are dynamically assigned a different IP address each time they login to the ISP server. The reason for the dynamic IP address assignment being that the ISP server has a limited number of IP addresses available, the number of IP addresses defined by the number of ports supported by the server. Therefore, as each new user logs into the ISP, the ISP assigns the user an IP address associated with the port they are currently occupying. Thus, unless a user happens to log into the same port each time, an option generally not within the discretion of the user, the user will not be assigned the same IP address.

Another problem resulting from the technical limitations of the prior art is that a home based computer system user, engaged on the internet from the only local PSTN extension within the home, cannot receive phone calls while the computer system is utilizing the PSTN extension. Heavy internet users have taken the only recourse currently available: the costly addition of another local PSTN extension within the home, wherein one of the extensions is primarily used for the computer system and the other primarily used for voice communications.

It will be appreciated by those skilled in the art, however, that the technical hurdle faced by the developers of the internet telephony applications is not trivial. To solve the problems associated with the prior art, an application will be required to seamlessly integrate the packet switched network of the internet with the circuit

switched network of the PSTN, a large undertaking which requires expertise in both the world of telecommunications and the Internet. Just such a solution is presented in the context of the present invention.

Thus a need exists for a method and apparatus for connecting an incoming call to a computer system that is already engaged in a communication session.

SUMMARY OF THE INVENTION

In accordance with the teachings of the present invention, a method and apparatus for connecting an incoming call to a computer system already engaged in an on-line communication session is disclosed. As illustrated in a first embodiment, a server, resident in a public switched telephone network (PSTN) to control communication services associated with a communication switch, includes at least one plain old telephone system (POTS) interface, a data network interface, a storage medium having a plurality of programming instructions stored thereon and an execution unit, coupled to the storage medium, the POTS interface and the data network interface, for executing the programming instructions. Wherein the plurality of programming instructions, when executed, implement a plurality of communication services including services to receive the telephone call from the communication switch at the POTS interface and interleavingly deliver incoming call signals to the client computer over the data network, and to accept interleaved outbound call signals from the client computer via the data network and to transmit such outbound call signals to an originating PSTN extension via the POTS interface.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be described by way of exemplary embodiments, but not limitations, illustrated in the accompanying drawings in which like references denote similar elements, and in which:

Figure 1 is a block diagram illustrating an exemplary communication system incorporating the teachings of the present invention;

Figures 2 and **3** are flow charts illustrating a method for connecting a phone call to a client computer already engaged in another communication session, in accordance with one embodiment of the present invention;

Figure 4 is a block diagram illustrating an adjunct processor incorporating the teachings of the present invention suitable for incorporation in the exemplary communication system of **Figure 1**;

Figure 5 is a block diagram illustrating the key software components of the adjunct processor of **Figure 4**, in accordance with one embodiment of the present invention;

Figure 6 is a block diagram of an alternate embodiment of a communication system incorporating the teachings of the present invention;

Figure 7 is a flow chart illustrating the method steps for connecting a phone call to a client computer already engaged in another communication session, in accordance with an alternate embodiment of the present invention; and

Figure 8 is a flow chart illustrating the method steps for connecting a phone call to a client computer already engaged in another communication session, in accordance with an alternate embodiment of the present invention.

DETAILED DESCRIPTION

In the following description, for purposes of explanation, specific numbers, materials and configurations are set forth in order to provide a thorough understanding of the present invention. However, it will be apparent to one skilled in the art that the present invention may be practiced without the specific details. In other instances, well known features are omitted or simplified in order not to obscure the present invention. Furthermore, for ease of understanding, certain method steps are delineated as separate steps, however, these separately delineated steps should not be construed as necessarily order dependent in their performance.

Referring now to **Figure 1**, a block diagram is presented illustrating an exemplary communication system **100** incorporating the teachings of the present invention for connecting an incoming call to a computer system that is already engaged in a communication session. As depicted, communication system **100** is shown comprising Adjunct Processor (AP) **112** incorporated with the teachings of the present

invention, internet telephony enabled client computer **102** and handset **140**. As will be described in more detail below, AP **112** and Internet Service Provide (ISP) **120** facilitate access by client computer **102** to one or more corporate presence servers (CPS) such as, for example CPS **155, 156, 157** and/or **159,** via Internet **150** and Public Switched Telephone Network (PSTN) **170** incorporated with the teachings of the present invention. In other words, ISP **120** and client computer **102** are engaged in a data communication session over communication lines **122** and **106**. Nevertheless, in accordance with the teaching of the present invention, AP **112** incorporated with the teachings of the present invention enables handset **140** to call internet telephony enabled client computer **102**, allowing a user of handset **140** and a user of client computer **102** to converse with each other, while client computer **102** remains connected to Internet **150**, and without requiring an additional communication line besides communication line **106**.

As illustrated, PSTN **170** is shown comprising Service Switching Points (SSP) **108** and **136**, Signal Transfer Points (STP) **126** and **130**, Signal Control Point (SCP) **134**, Adjunct Processor (AP) **112** incorporated with the teachings of the present invention and, in one embodiment, Internet Service Provider (ISP) **120**, coupled together as shown. In one embodiment, ISP **120** is integrated within SSP **108**, while in an alternate embodiment, ISP **120** may reside outside of the PSTN network **170**. In accordance with the illustrated example embodiment, client computer **102** communicates via PSTN **170** with a modulation/demodulation (MODEM) device coupled to PSTN extension **104**, as is well known in the art. PSTN extension **104** is coupled to SSP **108** with communication line **106**. As shown, SSP **108** may well be coupled to other extensions such as extension **142** via communication line **143**. In addition, SSP **108** is also coupled to STP **126** via trunk line **124**, and to AP **112** and ISP **120** via trunk lines **110** and **122**, respectively. Continuing, STP **126** is coupled to STP **130** via trunk line **128**, while STP **130** is coupled to SCP **134** and SSP **136** with control line **132** and trunk line **135**, respectively. As was the case with SSP **108**, SSP **136** is depicted as supporting a plurality of PSTN extensions such as extensions **140** and **144** via communication lines **138** and **145**, respectively.

As illustrated in **Figure 1**, extension **140** is intended to provide telephony service to a broad category of conventional handsets known in the art. Accordingly, extension **140** may well be referred to as handset **140**. That is, no special features are required of telephone handset **140** for it to call and be "connected" to client computer **102**. Additionally, in the embodiment of **Figure 1** communication line **138** connecting handset **140** to SSP **136** may simply be a plain old telephone service (POTS) communication line, although other types of communication lines may be used. Although the present invention is presented within the context of a PSTN communications system, those skilled in the art will appreciate that this is by example only, and that the teachings of the present invention may well be practiced in a wireless communication system such as, for example, cellular communication systems, Personal Communications Services (PCS) systems, and the like.

Client computer **102** is also intended to represent a broad category of internet telephony enabled computer systems known in the art. An example of such a computer system is a desktop computer system equipped with a high performance microprocessor, such as the Pentium® processor manufactured by Intel Corporation of Santa Clara, CA, a number of audio input and output peripherals/interface for inputting, digitizing and compressing outbound audio, and for decompressing and rendering inbound audio, a communication interface for sending and receiving various data packets (including audio data packets) in accordance with certain standard communication protocol, such as a V.42bis compliant modem, a windows-based operating system, such as Windows™ 95 developed by Microsoft Corporation of Redmond, WA, a web communications tool such as Navigator™, developed by Netscape Communications of Mountain View, CA, and an internet telephony application, such as the above described IPhone developed by Intel Corporation. Similarly, communication line **106** connecting client computer **102** to PSTN **170** may also be simply a POTS line, although other types of communication lines may also be used.

As depicted in **Figure 1**, AP **112** is coupled to ISP **120** via control line **118**, while ISP **120** is further coupled to SSP **108** via a plurality of other communication lines **123**. ISP **120** is coupled to Internet **150** via line **146**, while AP **112** is coupled to

Internet **150** via line **111**. Communication line **122** typically is of the same type as communication line **106**, e.g., POTS or integrated services digital network (ISDN) lines. Communication lines **111, 123** and **146** are intended to represent communication lines of various types, such as T1 (1.533 Mbps) trunk lines, or E1 (2.0488 Mbps) trunk lines which may be configured to serve client computers with higher data rate requirements, or to service a number of individual communication lines.

One skilled in the art of, for example, telecommunications, will appreciate that PSTN **170** is significantly more complex than what is depicted in **Figure 1**. For example, each SSP services thousands of extensions, and there are numerous SSPs, STPs and SCPs. However, **Figure 1** does capture a number of the more relevant components of PSTN **170** necessary to illustrate the interrelationship between PSTN **170**, client computer **102**, handset **104** and ISP **108**, such that one skilled in the art may practice the present invention. The functions performed by SSP **108** and **136**, STP **126** and **130** and SCP **134**, as well as their constitutions are well known. Accordingly, PSTN **170** will not be further described.

Similarly, it should be noted that while for ease of explanation only a handful of CPS **155-159** are shown to be connected to Internet **150**, there are actually numerous CPSs connected to Internet **150**, as well appreciated by those skilled in the art. In addition, there are government presence severs (not shown) as well as education presence servers (not shown) coupled to Internet **150** as is well known in the art, wherein each of these servers may be directly connected to Internet **150** as illustrated, or indirectly through an ISP such as ISP **158**. Also, while the present invention is being described in the context of client computer **102** being engaged in data communication with ISP **120**, as will be readily apparent from the description to follow, the present invention may be practiced with any client computer **102** engaged in data communication with any server.

Turning now to **Figures 2** and **3**, a flow chart illustrating the method steps of one example for connecting a call from handset **140** to client computer **102** is shown. As depicted, the method begins at step **204** when a user of telephone handset **140** initiates a telephone call by dialing PSTN extension **104**. As described earlier, the call is established via communication line **138** through SSP **136**, STPs **130** and **126** to SSP

108. In particular, as the call is placed from telephone handset **140** via SSP **136**, SSP **136** will encode Automatic Number ID (ANI) information associated with telephone handset **140** and Destination Number ID Service (DNIS) information associated with the call destination (i.e., PSTN extension **104**) in the prompt signal from SSP **136** to SSP **108**, in an attempt to initiate the telephone call. In step **206**, SSP **108** detects the inbound call destined for PSTN extension **104** from telephone handset **140** via SSP **136**, and determines that PSTN extension **104** is currently in use. Accordingly, in step **208**, SSP **108** queries AP **112** via trunk line **110** to determine the calling features associated with PSTN extension **104** to determine possible alternate call routing options for the inbound call. To determine the calling features associated with PSTN extension **104**, AP **112** accesses database **114** which maintains a list of calling features associated with all PSTN extensions serviced by SSP **108**. One such feature, in accordance with the teachings of the present invention, is feature indicating that a particular PSTN extension is internet telephony enabled. Accordingly, in step **210**, a determination is made by AP **112** as to whether PSTN extension **104** is an internet telephony enabled PSTN extension.

If in step **210**, it is determined that PSTN extension **104** is not an internet telephony enabled extension, AP **112** continues down the calling feature hierarchy associated with PSTN extension **104**, in step **212**, to determine how to best route the call. Examples of such alternative calling features may include call waiting, or voice mail, to name but a few. In step **214**, based on input from AP **112**, SSP **108** routes the inbound call in accordance with the calling features associated with PSTN extension **104**, as appropriate, and the method ends in step **216**.

If, however, in step **210**, it is determined that PSTN extension **104** is, in fact, internet telephony enabled, SSP **108** queries ISP **120** to determine if ISP **120** has the capacity available to accept and route the inbound call to PSTN extension **104**. In step **220**, a determination is made by ISP **120** as to whether ISP **120** can route the call to PSTN extension **104**. In making the determination of step **220**, ISP **120** first determines whether it is currently engaged in a data communication session with client computer **102** via PSTN extension **104**, and also whether there are available communication ports with which to accept the incoming call. If, in step **220**, it is

determined that ISP **120** cannot accept and route the inbound call, then the method continues with step **212**, wherein AP **112** continues down the calling feature hierarchy associated with PSTN extension **104** to determine how to route the call.

If, alternatively, it is determined in step **220** that ISP **120** has the resources available to accept and route the inbound call, ISP **120** instructs SSP **108** to forward the inbound call to ISP **120** via an available communication line **123**, in step **302**. In step **304**, ISP **120** signals client computer **102** via communication line **122**, SSP **108** and communication line **106** supporting the data communication session with Internet **150**, of the inbound call and providing additional information associated with the origin (ANI), format and any toll charges associated with accepting and completing the inbound call, and queries the user of client computer **102** whether or not to accept the inbound call. In step **306**, the user of client computer **102** responds to ISP **120** via the data communication session with Internet **150** (i.e., supported by communication line **106**, SSP **108** and communication line **122**). If, in step **306**, the user of client computer **102** chooses not to accept the inbound call, or if the prompt from ISP **120** goes unanswered for a predetermined period of time, the method continues with step **211**, wherein one embodiment control transfers back from ISP **120** to AP **112**, which continues down the calling feature hierarchy associated with PSTN extension **104** to determine how best to route the inbound call.

If, however, in step **306** the user of client computer **102** signals ISP **120** to accept and route the inbound call, ISP **120** completes a communication connection between communication line **123** and communication line **122** acting as a "bridge" between handset **140** and internet telephony enabled client computer **102**, in step **308**. In particular, ISP **120** digitizes and compresses inbound call signals received from handset **140** on communication line **123** and interleavingly delivers the encoded call signals to client computer **102** via communication line **122**, SSP **108** and communication line **106**, or, in other words, via the previously established data connection. The compressed inbound call signals will be decompressed by the communication interface of client computer **102** and rendered by the internet telephony application. Likewise, outbound call signals emanating from client computer **102** will be digitized by the audio interface and compressed by the communication interface of

client computer **102** and interleavingly delivered to ISP **120** wherein they will be decompressed and rendered for the benefit of the user of telephone handset **140**. In other words, ISP **120** converts the voice information between PSTN and IP protocols and interleaving delivers call signals to/from telephone handset **140** and client computer **102** until call completion, and the method ends in step **310**.

Jumping ahead to **Figure 7**, a flow chart illustrating the method steps of an alternate embodiment of the present invention for connecting a call from handset **140** to client computer **102** is shown. As depicted, the method begins with step **702**, wherein the user of client computer **102**, in anticipation of beginning an internet data communication session via PSTN extension **104**, actively enables a call forwarding feature with the switch of SSP **108** by entering a feature_enable key sequence (e.g., *71) using the keypad of the telephone connected to PSTN extension **104**. In response, the switch of SSP **108** prompts the user of PSTN extension **104** to enter the telephone number to which calls are to be forwarded, wherein the user enters an access number corresponding to ISP **120** using the telephone keypad, for example. In an alternate embodiment the feature_enable key sequence and entry of the access number for ISP **120** may be performed automatically, as part of a modem initialization script at the onset of a data communications session. In the context of the present example implementation, the user of client computer **102** then dials ISP **120** to establish a data connection with a server (e.g., CPS **156**) through Internet **150**.

Having actively enabled the call forwarding feature in step **702**, PSTN extension **104** is now enabled to receive inbound calls via ISP **120**. In step **704**, a user of telephone handset **140** initiates a telephone call by dialing PSTN extension **104**. As described earlier, the call is established via communication line **138** through SSP **136**, STPs **130** and **126** to SSP **108**. In particular, as the call is placed from telephone handset **140** via SSP **136**, SSP **136** will encode Automatic Number ID (ANI) information associated with telephone handset **140** and Destination Number ID Service (DNIS) information associated with the call destination (i.e., PSTN extension **104**) in the prompt signal from SSP **136** to SSP **108**, in an attempt to initiate the telephone call. In step **706**, SSP **108** receives the incoming call request from SSP **136**, determines that PSTN extension **104** denoted by DNIS has forwarded all incoming calls to an access

number corresponding to ISP **120**. Accordingly, SSP **108** attempts to forward the call to ISP **120**. In one embodiment, the original DNIS corresponding to PSTN extension **104** becomes the ANI for the forwarded call, while the access number corresponding to ISP **120** becomes the new DNIS for the forwarded call. In an alternate embodiment, all three numbers (i.e., the original ANI, the original DNIS, and the access number corresponding to ISP **120**) may be retained. If, in step **708**, it is determined that ISP **120** does not have the necessary resources (or, capacity) available to route the incoming call to PSTN extension **104**, ISP **120** rejects the incoming call from SSP **108**, and SSP **108** continues down the calling feature hierarchy associated with PSTN extension **104** to route the call, or simply generates a "busy" tone, in step **710**.

If, however, in step **708** it is determined that ISP **120** has the resources available to accept and route the incoming call to PSTN extension **104**, ISP **120** determines that the ANI information received corresponds to a particular client computer (e.g., client computer **102**), using one of a number of methods known in the art (e.g., caller ID services), and signals client computer **102** of the incoming call. In so doing, ISP **120** prompts the user of client computer **102** with associated call information and queries the user whether to accept the incoming call, in step **714**. If, in step **716**, the user denies the incoming call, ISP **120** rejects the incoming call from SSP **108**, and SSP **108** continues down the calling feature hierarchy associated with PSTN extension **104** to route the call, or simply generates a "busy" tone, in step **710**.

Alternatively, if the user of client computer **102** chooses to accept the call in step **716**, ISP **120** completes a communication connection between communication line **123** and communication line **122** acting as a "bridge" between handset **140** and internet telephony enabled client computer **102**, in step **718**. In particular, ISP **120** digitizes and compresses inbound call signals received from handset **140** on communication line **123** and interleavingly delivers the encoded call signals to client computer **102** via communication line **122**, SSP **108** and communication line **106**, or, in other words, via the previously established data connection. The compressed inbound call signals will be decompressed by the communication interface of client computer **102** and rendered by the internet telephony application. Likewise, outbound call signals emanating from client computer **102** will be digitized by the audio interface and compressed by the

communication interface of client computer **102** and interleavingly delivered to ISP **120** wherein they will be decompressed and rendered for the benefit of the user of telephone handset **140**. That is to say, ISP **120** converts the voice information between PSTN and IP protocols and interleaving delivers call signals to/from telephone handset **140** and client computer **102** until call completion, and the method ends in step **720**.

Once the internet data connection is completed, the user of client computer **102** enters a feature_disable key sequence (e.g., *72) using the telephone handset connected to PSTN extension **104** disabling the call forwarding feature at the switch of SSP **108**, thereby enabling PSTN extension **104** to receive telephone calls as normal. In alternate embodiments, the feature_disable key sequence may also be implemented as part of a modem disconnection sequence, thereby relieving the user of client computer **102** with having to remember to disable the call forwarding feature.

Thus, for the illustrated embodiment of **Figure 1**, ISP **120** enables a user of handset **140** to call the user of client computer **102** by dialing the telephone number for extension **104**, and converse with the user of client computer **102**, even when user of client computer **102** is already using extension **104** to support a data connection with Internet **150**. In addition to allowing the user of client computer **102** to receive incoming calls from a remote telephone handset (e.g., handset **140**), the present invention contemplates that ISP **120** may also incorporate a number of additional value added features such as voice mail, caller identification (via ANI) and the ability to recognize and display the originator of the incoming call (e.g., from within a PBX system) as well as the call type (i.e., voice, fax, video, etc.) with the ability to route the call to an alternate location that is best-suited to handle the incoming call.

In addition, it will be recognized by those skilled in the art that the number of calls received by the user of client computer **102** is limited only by the number of extensions connected to communication lines **123**, and the bandwidth available on the communication connection between client computer **102** and ISP **120**. Therefore, if client computer **102** employs a 28.8 kbps modem to support the data connection to ISP **120**, and a full duplex voice connection typically consumes between 5 and 10 kbps, it is possible to support multiple voice connections within the 28.8 kbps bandwidth constraint imposed by the modem, for example. Thus, in one embodiment of the

present invention, ISP **120** may support several simultaneous (e.g., conferencing) voice connections to the user of client computer **102** within the bandwidth constraint of the modem, wherein the user of client computer **102** may simultaneously communicate with all of the callers (provided that the internet telephony application resident on client computer **102** supports such a conferencing feature). In alternate embodiments, where the available bandwidth or internet telephony software resident on client computer **102** cannot not support multiple simultaneous voice connections, ISP **120** may receive and support multiple calls to the user of client computer **102,** where the user of client computer **102** must selectively communicate with individual calls by placing the other(s) on hold.

Turning next to **Figure 8**, a flow chart illustrating the method steps of yet another example embodiment of the present invention is depicted. In accordance with illustrated example embodiment of **Figure 8**, AP **112** incorporating the teachings of the present invention is operative to receive an indication of an incoming call destined for PSTN extension **104**, engaged in a data communication session by client **102**, accept the call from the originating PSTN extension and interleavingly route the telephone call to the client computer via the data network. In one embodiment, the data network is Internet **150**, wherein AP **112** is communicatively coupled with Internet **150** via trunk line **111**. Thus, in accordance with the teachings of the present invention, to be described more fully below, AP **112** endowed with the teachings of the present invention is able to route the incoming call to client computer **102** without the need of an ISP's bridge gateway.

As in the example embodiments above, the method begins as SSP **108** receives an incoming call request destined for PSTN extension **104**, and detects that extension **104** is already in use, step **804**. Accordingly, SSP **108** queries AP **112** to identify the calling features associated with PSTN extension **104** to facilitate call routing, step **806**. In step **808**, AP determines whether extension **104** is engaged in a data communication session. That is, AP **112** determines whether extension **104** is being used by client computer **102** in a data communication session over a data network to which AP **112** has access. In the context of **Figure 1**, for example, client computer **102** engaged in a data communication session over Internet **150** is accessible via AP **112** via trunk line

111. As alluded to above, one method for making such a determination is to have client computer **102** register with AP **112** as the data session begins. Those skilled in the art will appreciate, however, that a number of alternative methods exist for determining the nature the telephone call.

If, in step **808**, it is determined that PSTN extension **104** is not engaged in a data communication session, e.g., it is being used for a voice conversation, or a facsimile transmission, etc., AP **112** continues down the calling feature hierarchy, described above, in determining how best to handle the call, step **810**. In step **812,** based at least in part on input from AP **112**, SSP **108** terminates the call (voice mail, busy signal, etc.) in accordance with the selected call handling feature from the hierarchy of call handling features, whereafter, the call ends at block **814**.

If, however, AP **112** determines in step **808** that PSTN extension **102** is engaged by computer **102** in a data communication session, AP **112** determines whether client computer **102** is internet telephony enabled, step **816**. If it is determined in step **816** that client computer **102** is not internet telephony enabled, the process continues with step **810** as AP **112** identifies an appropriate calling feature from the hierarchy of calling features associated with PSTN extension **104**.

Alternatively, in accordance with one aspect of the present invention, if client computer **102** is internet telephony enabled, AP **112** endowed with the teachings of the present invention signals client computer **102** of the incoming call with associated information (e.g., telephone number of originating extension, etc.) and queries client computer **102** of whether to accept the call, step **818**. If client **102** elects to accept the call, step **820**, AP **112** incorporating the teachings of the present invention directs SSP **108** to route the telephone call to AP **112**, which converts the voice to/from PSTN to IP protocol and interleavingly delivers the call signals to/from the orignating PSTN extension and client computer **102** via the data network employed for the data communication session and the PSTN network until call completion, step **822**. In accordance with the illustrated example embodiment of **Figure 1**, for example, AP **112** accepts the telephone call from SSP **108**, and interleavingly delivers call signals to/from client computer **102** over Internet **150** via trunk lines **111** and **146**, ISP **120**, SSP **108** and POTS line **106**, as shown.

Having presented a number of alternative embodiments above, attention is directed to **Figures 4** and **5**, which illustrate block diagrams of the hardware and software elements of exemplary computer server **400**, suitable for use as ISP **120** incorporating the teachings of the present invention, AP **112** incorporating the teach0ings of the present invention (or, in alternate embodiments, an AP/ISP combination). As illustrated in **Figure 4**, exemplary computer server **400** is comprised of multiple processors **402a - 402n** and memory subsystem **408** coupled to processor bus **404** as depicted. Additionally, computer server **400** is comprised of a second bus **412** and a third bus **410**. In one embodiment of the present invention bus **412** is a Peripheral Component Interconnect (PCI) bus **412**, while bus **410** is an Industry Standard Architecture (ISA) bus **410**. PCI bus **412** and ISA bus **410** are bridged to processor bus **404** by I/O controller **406**. Coupled to ISA bus 410 are display **414**, keyboard and cursor control device **424** and mass storage device **422** (e.g., hard drive). Additionally, exemplary computer server **400** is shown comprising PSTN interface **416** and Signaling System 7 (SS7) Interface **418**.

As depicted in **Figure 4**, PSTN interface **416** provides the necessary hardware to interface exemplary computer server **400** to a plurality of PSTN communication lines (e.g., T1, E1 or POTS), wherein the actual number of PSTN communication lines interfaced will be implementation dependent. Additionally, PSTN interface **416** provides advanced DSP-based voice, dual-tone multiple frequency (DTMF) and call progress functionality, which allows for downloadable DSP protocol and voice processing algorithms, thereby providing CODEC support locally on the interface. Examples of supported codecs include the Global System for Mobile Communications (GSM) codec and the ITU-T G.723.1 protocol codecs, the specification for which are commonly available from the GSM consortium and the International Telecommunications Union, respectively. Similarly, SS7 interface **418** provides the hardware necessary to interface exemplary computer server **400** with PSTN trunk lines (e.g., ISDN) supporting the out-of-band communication protocol (e.g., Signaling System 7 (SS7)) used between PSTN network elements (i.e., SSP-SSP, SSP-STP, STP-SCP, etc.). In one embodiment of the present invention PSTN interface **416** is preferably an AG-T1™ (for U.S. implementations, while an AG-E1 may be seamlessly

substituted for European implementations), while SS7 interface **418** is preferably the TX3000™, both of which, along with their accompanying software drivers, are manufactured by and commonly available from Natural MicroSystems of Natick, Massachusetts. Otherwise, all other elements, processors **402**, memory system **408** and so forth perform their conventional functions known in the art. Insofar as their constitutions are generally well known to those skilled in the art, they need not be further described.

From a software perspective, **Figure 5** illustrates the software elements of exemplary computer server **400**. In particular, exemplary computer server **400** is shown comprising an application layer consisting of Hop-on/Hop-off driver **502** and interapplication manager (Management Driver) **504**. Hop-on/Hop-off driver **502** and Management Driver **504** implements the method steps of **Figures 2**, **3**, **7** and **8** using services abstracted through SAL **506**.

The Service Abstraction Layer (SAL) **506** is shown comprising Signaling System 7 (SS7) Services **508**, Telephony Services **510**, Streaming Services **512**, Connection Services **514** and Data Services **516**. The interface layer is shown comprising Telephony Application Programming Interface (API) **520**, PSTN Data interface **522** and Connection/Compatibility Interface **524**. The driver layer is shown comprising SS7 driver **518**, PSTN driver **526** and Winsock/2 standard drivers (e.g., TCP/IP, etc.) **528**, all of which exchange information in the fashion depicted in **Figure 5**.

Within the context of the present invention, one purpose of SAL **506** is to provide and Application Programming Interface (API) for all the available services in exemplary computer server **400**. The API will abstract out the actual modules used for providing services such as connection establishment (514), streaming and data exchange services (512, 516). Additionally, SAL **506** provides the common operation tools such as queue management, statistics management, state management and the necessary interface between the plug-in services (i.e., drivers in the driver layer). SAL **506** is also responsible for loading and unloading the appropriate drivers as appropriate.

Streaming service **512** is responsible for interfacing with the components that provide the real-time streaming functionality for the multimedia data. Once the

connection has been established between the connection points (i.e., PSTN, H.323 or Wave files), streaming service **512** will take over the management and streaming of data between the two connected parties, until the connection is terminated. Similarly, data service layer **516** is responsible for providing non real-time peer to peer (i.e., computer-computer) messaging and data exchange between exemplary computer server **400** and other Internet and perhaps PSTN based applications. Sending messages to exemplary to computer server end-points (i.e., other similarly equipped ISPs on the Internet) or other servers within the PSTN is accomplished via data service layer **516**.

Connection service **514** works in conjunction with streaming services **512** and data services **516**. In particular, connection service **514** includes a connection establishment and tear-down mechanism facilitating the interconnection to the PSTN. Additionally, for the illustrated embodiment, connection service **514** employs connection and compatibility services **524** which facilitate interoperation between communication equipment that support industry standards, thereby allowing a variety of communication equipment manufactured by different vendors to be benefited from the present invention. Connection and compatibility services **524** include, in particular, services for supporting standard video telephony, such as ITU-T's H.323 video telephony, and standard data communication, such as ITU-T's T.120 data communication protocol. Examples of the connection establishment and tear-down mechanisms supported by connection service layer **514** include opening and starting PSTN ports, call control, DTMF collection, and tone generation, to name but a few.

As alluded to with respect to **Figure 4**, SS7 driver **518** and PSTN driver **526** serve to integrate SS7 interface **418** and PSTN interface **416** with the other features and functions of exemplary computer server **400**. As illustrated in **Figure 5**, SS7 driver **518** communicates with SS7 service layer **508**, while PSTN driver **526** communicates with telephony API **520** and PSTN data API **522**.

Telephony services **510** service all communications with client computers, such as client computer **102** using telephony API **520** and PSTN driver **526**. Telephony services **510** include in particular services for handling computer telephony **506**, and automatic call distribution (ACD) necessary for Private Branch Exchange (PBX) based systems. PSTN driver **526** is equipped to facilitate the above described compression

and transmission of inbound call signals from handset **140** as well as decompression and transmission of outbound call signals from client computer **102**. PSTN driver **526** supports these functions with the necessary interfaces to accommodate both trunk line and POTS communication line interfaces common to SSP **108**. Implementation of these services, as described above, is well within the ability of those skilled in the art of, for example, telecommunications.

Although depicted as separate entities in **Figure 1**, those skilled in the art will appreciate that AP **112** and ISP **120** are both, essentially, computer servers and may, in one embodiment be the same computer server. In such an embodiment, communication line **118** represents a communication bus internal to the combined AP/ISP server. Alternatively, AP **112** and ISP **120** may be co-located in a local office of PSTN **170** as part of a common Local Area Network (LAN), where in such an embodiment communication line **118** represents a LAN communication path such as, for example, an Ethernet communication line. In yet another alternative embodiment, AP **112** may be located in a local office (not shown) of PSTN **170**, while ISP **120** is remotely located, wherein such instance communication line **118** represents a digital control line to AP **112** allowing for out-of-band signaling communication.
An example of such a communication system architecture is illustrated in **Figure 6.**

In particular, **Figure 6** illustrates a block diagram of an alternative communication system architecture incorporating the teachings of the present invention. As illustrated, the elements of communication system **600** are much the same as the architecture depicted in exemplary communication system **100**, with the primary distinction arising from the remote placement of ISP **120**. In particular, in communication system **600**, ISP **120** and AP **112** are incorporated with the teachings of the present invention, however, communication between these network elements must occur via a PSTN trunk line **602**. An example of a trunk line suitable for use as trunk line **602** is a primary rate ISDN (PRISDN) line, well known to those skilled in the art. Except for the location of ISP **120** relative to AP **112**, the operation of the present invention is much the same as that previously described.

Thus, a method and apparatus for connecting an incoming call to a computer system that is already engaged in a communication session has been described. While

the internet telephony call feature of the present invention has been described in terms of the above illustrated embodiments, those skilled in the art will recognize that the invention is not limited to the embodiments so described. For example, the Public Switched Telephone Network and the Internet are complex communication systems and, therefore, a number of alternative variations of the present invention may be practiced without deviating from the spirit and scope of the present invention. Further, although the exemplary embodiments were presented in the context of the PSTN, one skilled in the art will appreciate that the spirit and scope of the present invention lends itself equally well to the world of wireless communications. For example, the present invention anticipates that the originator of the telephone call could be call from a cellular telephone; or that the client computer could be coupled to a wireless communications network via a wireless communications modem over a wireless communication channel, in which case, in accordance with the present invention, the telecomputing session would not necessarily prohibit the receipt of incoming cellular calls to the wireless handset. Thus, the present invention may be practiced with modification and alteration within the spirit and scope of the appended claims. Accordingly, the description is to be regarded as illustrative instead of restrictive on the present invention.

CLAIMS

We claim:

1. A server, resident in a public switched telephone network (PSTN), to control communication services associated with a communication switch, the server comprising:

at least one plain old telephone system (POTS) interface, to interface with the PSTN;

a data network interface coupled to a data network including a client computer; a storage medium having stored therein a plurality of programming instructions; and an execution unit, coupled to the storage medium, for executing at least a subset of the programming instructions to implement communication services to route a telephone call destined for a PSTN extension to the client computer which has engaged the PSTN extension for a communication session over the data network, the communication services including services to receive the telephone call from the communication switch at the POTS interface and interleavingly deliver incoming call signals to the client computer over the data network, and to accept interleaved outbound call signals from the client computer via the data network and to transmit such outbound call signals to the PSTN extension via the POTS interface.

2. The server of claim 1, wherein the communication services are executed without any disruption to the data communication session of the client computer over the data network.

3. The server of claim 1, wherein the server is an adjunct processor (AP) to provide the PSTN with a hierarchy of calling features, including communication services for routing a telephone call over the data network.

4. The server of claim 3, wherein the hierarchy of calling features include one or more of internet protocol (IP) telephony calling features, voice mail calling features, call waiting calling features, and the like.

5. The server of claim 1, wherein the data network interface is operative to interface the server with an global communications network.

6. The server of claim 5, wherein the global communications network is the

Internet.

7. The server of claim 1, wherein the data network interface is operative to interface the server with an Intranet.

8. The server of claim 1, wherein the communication services include bridging a circuit switched network and a packet switched network to route a telephone call originating on the circuit switched network to the client computer on the packet switched network, wherein the client computer is already engaged in a data communication session on the packet switched network and the communication services fail to disrupt the data communication session.

9. The server of claim 1, further comprising an audio interface codec for encoding incoming call signals destined for the client computer, and for decoding outbound call signals destined for the originating PSTN extension.

10. The server of claim 1, wherein the communication services include services to decode caller identification information from the incoming telephone call.

11. The server of claim 1, wherein the communication services further include services for packing the incoming call signals into packets for delivery to the client computer, and unpacking the outbound call signals for transmission to the originating PSTN extension.

12. A method for connecting an incoming call from an originating public switched telephone network (PSTN) extension to a destination PSTN extension, wherein the destination PSTN extension is engaged by a client computer for a data communication session, the method comprising:

(a) detecting the incoming call at a communication switch associated with the PSTN;

(b) signalling a server resident within the PSTN and associated with the communication switch of the incoming call;

(c) answering the incoming call at the server on behalf of the destination PSTN extension and interleavingly delivering incoming call signals received from the PSTN to the client computer over a data network, and accepting interleaved outbound call signals from the client computer via the data network

and transmitting the outbound call signals to the originating PSTN extension via the PSTN.

13. The method of claim 12, wherein the server determines whether the destintion PSTN extension is engaged in a data communication session and, if so, whether the client computer is internet protocol (IP) telephony enabled.

14. The method of claim 14, wherein server is an adjunct processor (AP) associated with the communication switch which bridges telephone calls between the PSTN and the data network to provide a heirarchy of communication services including internet protocol (IP) telephony.

15. An Adjunct Processor (AP), coupled to a Service Switching Point (SSP) in a public switched telephone network (PSTN), to connect an incoming call to a client computer, wherein the incoming call is initiated by a telephone handset bound for a PSTN extension currently engaged by a client computer in a data communication session with a server on a public computer network, the AP comprising:

a storage medium having stored therein a plurality of programming instructions; and

an execution unit, coupled to the storage medium, for executing the programming instructions to implement communication services for facilitating the connection of the incoming call to the client computer, wherein the communication services include services for answering the incoming call on behalf of the client computer and interleavingly delivering incoming call signals to the client computer, and accepting interleaved outbound call signals from the client computer and transmitting the outbound call signals on behalf of the client computer to the telephone handset, as well as programming instructions integrating the AP with the public computer network.

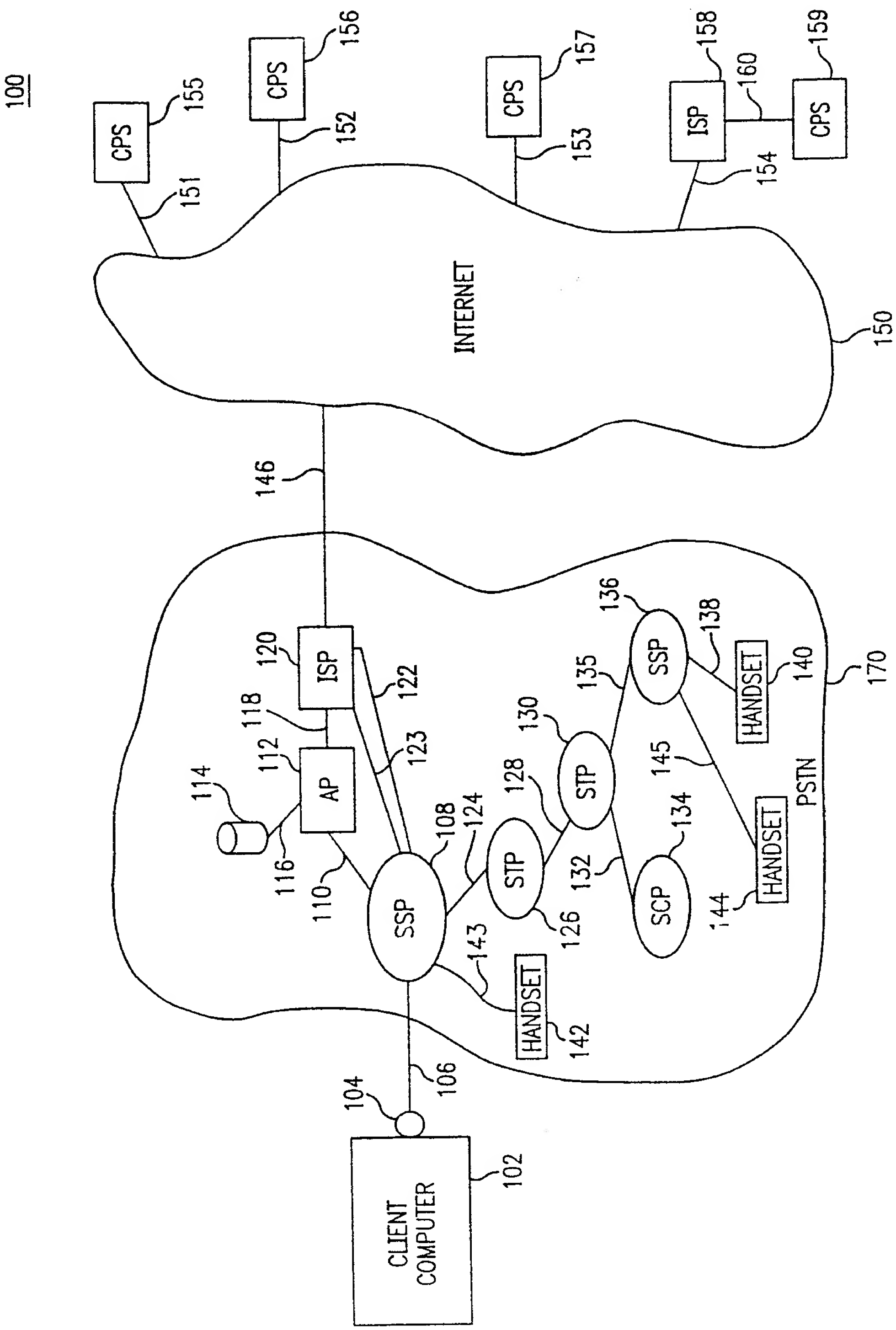

100
CPS
155
151
CPS
156
152
CPS
157
153
ISP
158
160
CPS
159
154
INTERNET
150
146
ISP
120
118
122
123
AP
112
114
116
110
SSP
108
124
STP
126
128
STP
130
132
SCP
134
135
SSP
136
138
HANDSET
140
145
HANDSET
144
143
HANDSET
142
PSTN
170
106
104
CLIENT COMPUTER
102

FIG. 1

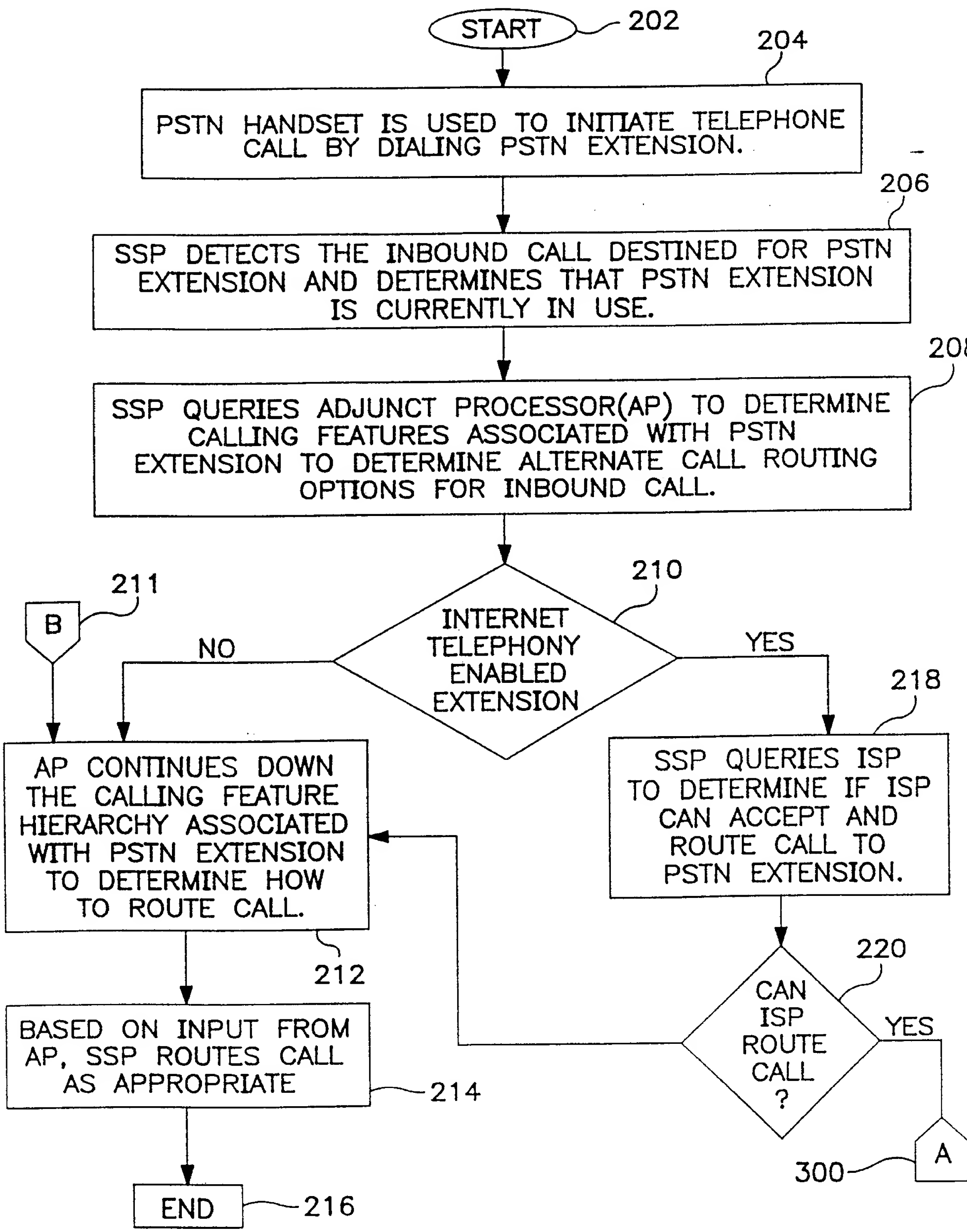

START
202
204
PSTN HANDSET IS USED TO INITIATE TELEPHONE CALL BY DIALING PSTN EXTENSION.
206
SSP DETECTS THE INBOUND CALL DESTINED FOR PSTN EXTENSION AND DETERMINES THAT PSTN EXTENSION IS CURRENTLY IN USE.
208
SSP QUERIES ADJUNCT PROCESSOR(AP) TO DETERMINE CALLING FEATURES ASSOCIATED WITH PSTN EXTENSION TO DETERMINE ALTERNATE CALL ROUTING OPTIONS FOR INBOUND CALL.
211
B
210
INTERNET TELEPHONY ENABLED EXTENSION
NO
YES
218
SSP QUERIES ISP TO DETERMINE IF ISP CAN ACCEPT AND ROUTE CALL TO PSTN EXTENSION.
AP CONTINUES DOWN THE CALLING FEATURE HIERARCHY ASSOCIATED WITH PSTN EXTENSION TO DETERMINE HOW TO ROUTE CALL.
212
220
CAN ISP ROUTE CALL ?
YES
BASED ON INPUT FROM AP, SSP ROUTES CALL AS APPROPRIATE
214
A
300
END
216

FIG. 2

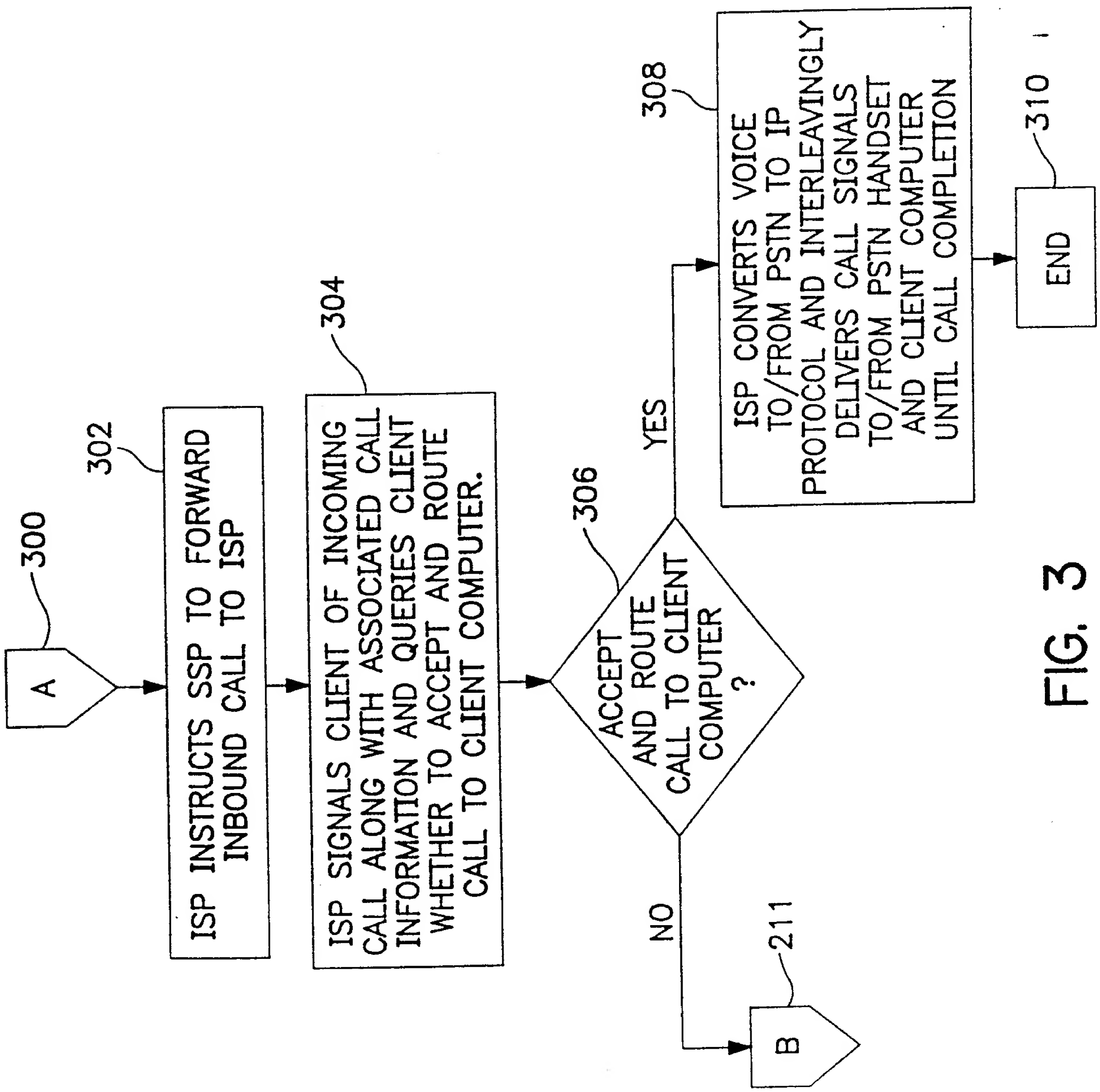

A
300
302
ISP INSTRUCTS SSP TO FORWARD INBOUND CALL TO ISP
304
ISP SIGNALS CLIENT OF INCOMING CALL ALONG WITH ASSOCIATED CALL INFORMATION AND QUERIES CLIENT WHETHER TO ACCEPT AND ROUTE CALL TO CLIENT COMPUTER.
306
ACCEPT AND ROUTE CALL TO CLIENT COMPUTER ?
YES
NO
308
ISP CONVERTS VOICE TO/FROM PSTN TO IP PROTOCOL AND INTERLEAVINGLY DELIVERS CALL SIGNALS TO/FROM PSTN HANDSET AND CLIENT COMPUTER UNTIL CALL COMPLETION
310
END
211
B

FIG. 3

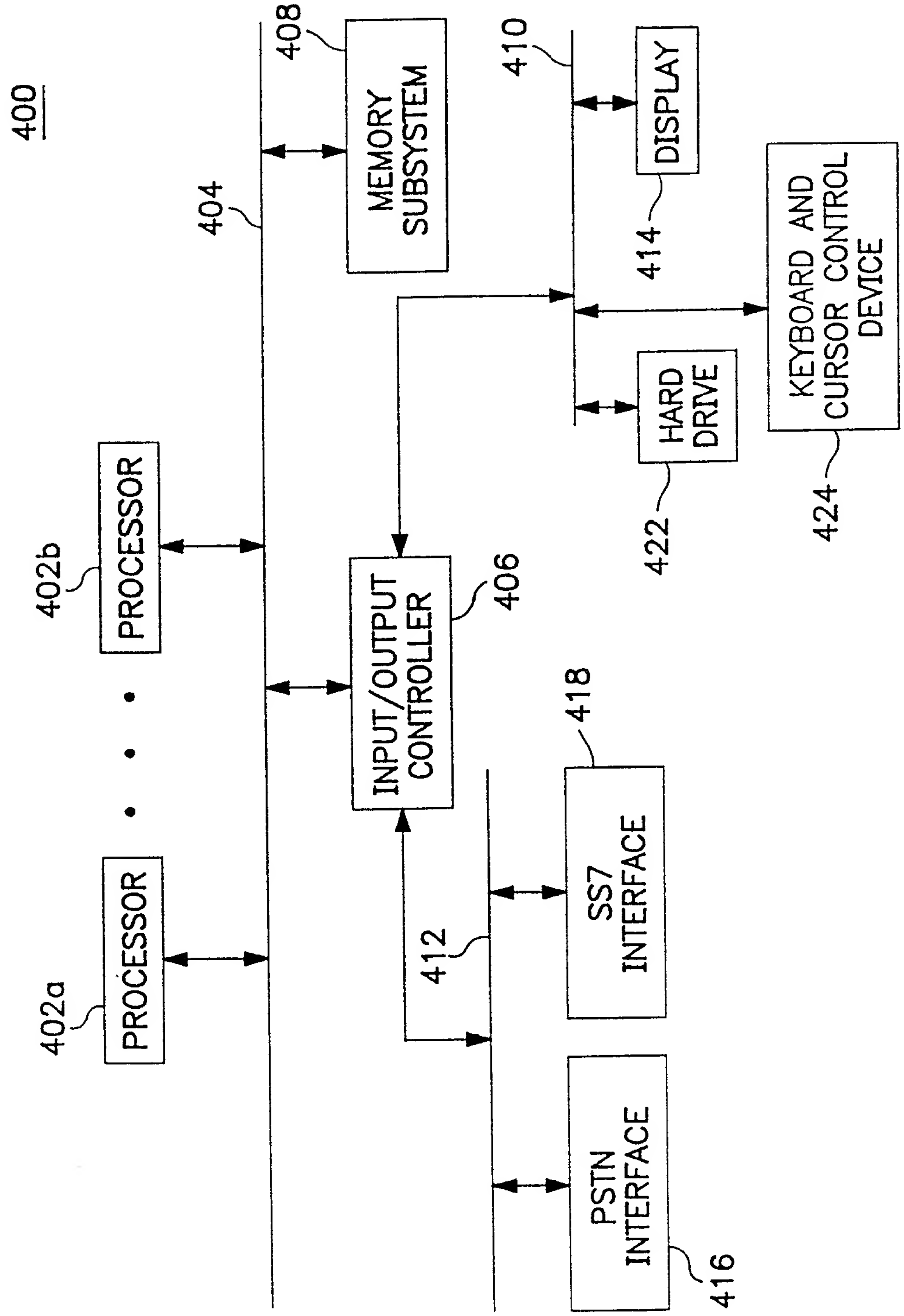

400
402a
PROCESSOR
402b
PROCESSOR
404
408
MEMORY SUBSYSTEM
INPUT/OUTPUT CONTROLLER
406
410
DISPLAY
414
HARD DRIVE
422
KEYBOARD AND CURSOR CONTROL DEVICE
424
412
SS7 INTERFACE
418
PSTN INTERFACE
416

FIG. 4

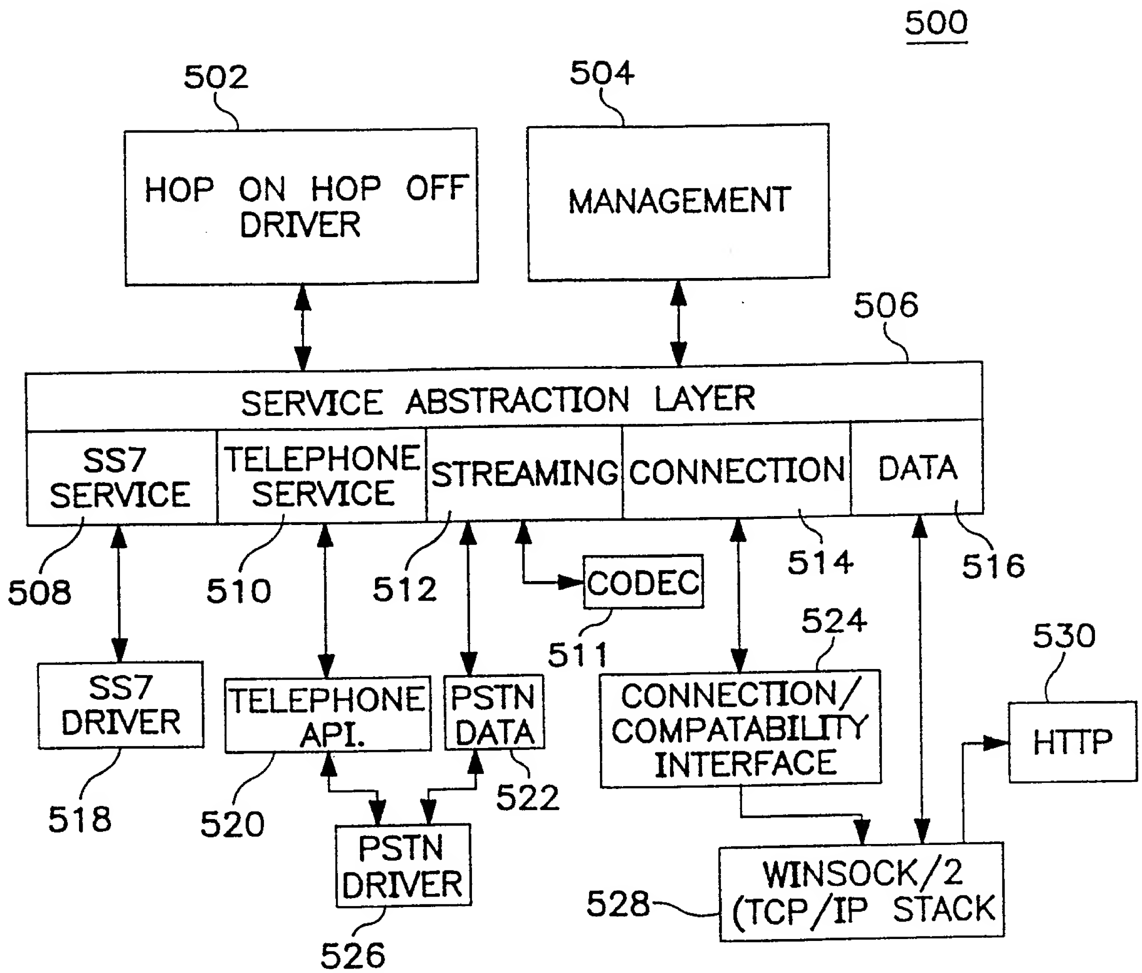

500
502
HOP ON HOP OFF DRIVER
504
MANAGEMENT
506
SERVICE ABSTRACTION LAYER
SS7 SERVICE
TELEPHONE SERVICE
STREAMING
CONNECTION
DATA
508
510
512
514
516
CODEC
511
524
530
SS7 DRIVER
TELEPHONE API.
PSTN DATA
CONNECTION/ COMPATABILITY INTERFACE
HTTP
518
520
522
PSTN DRIVER
526
528
WINSOCK/2 (TCP/IP STACK

FIG. 5

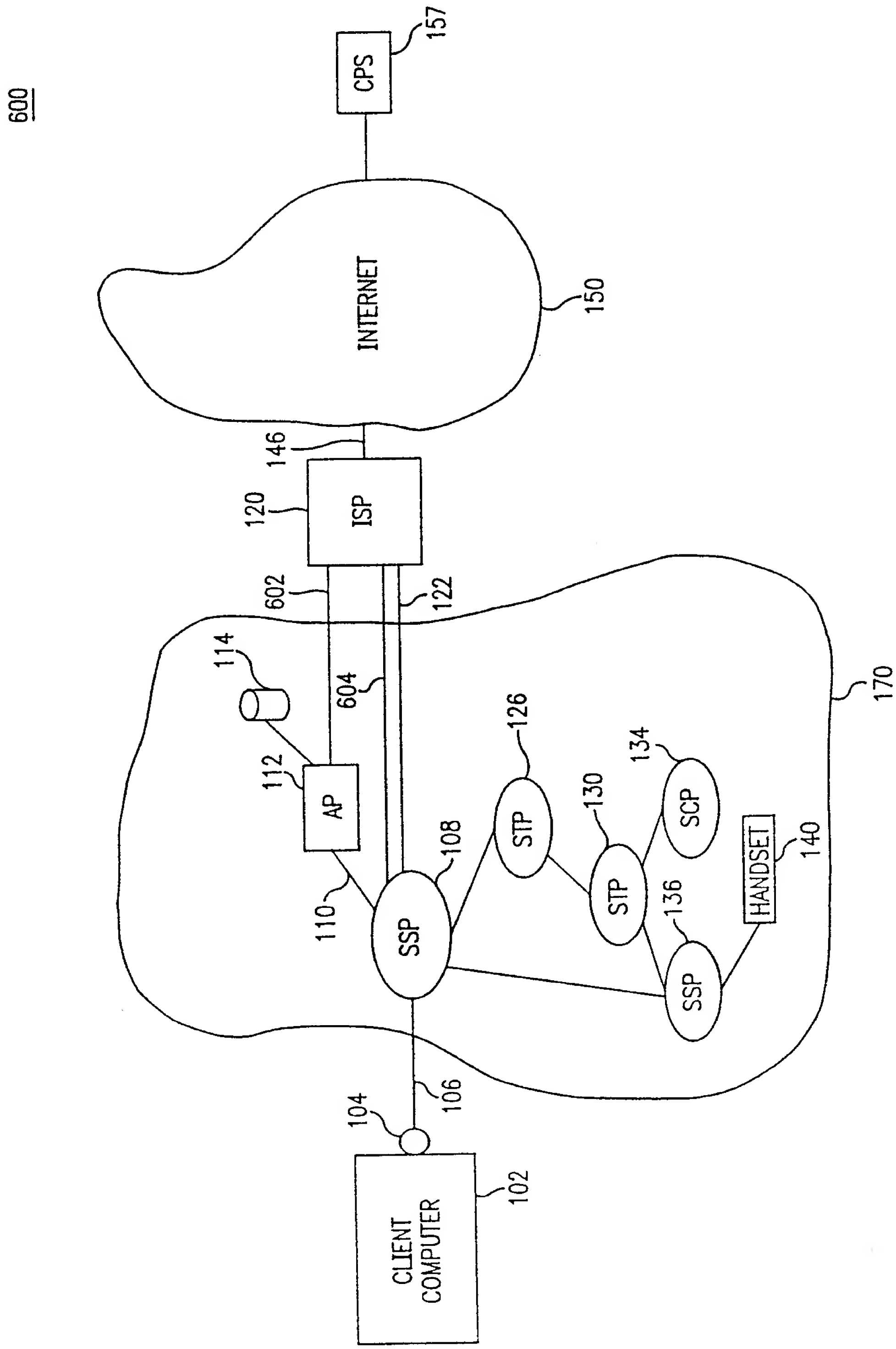
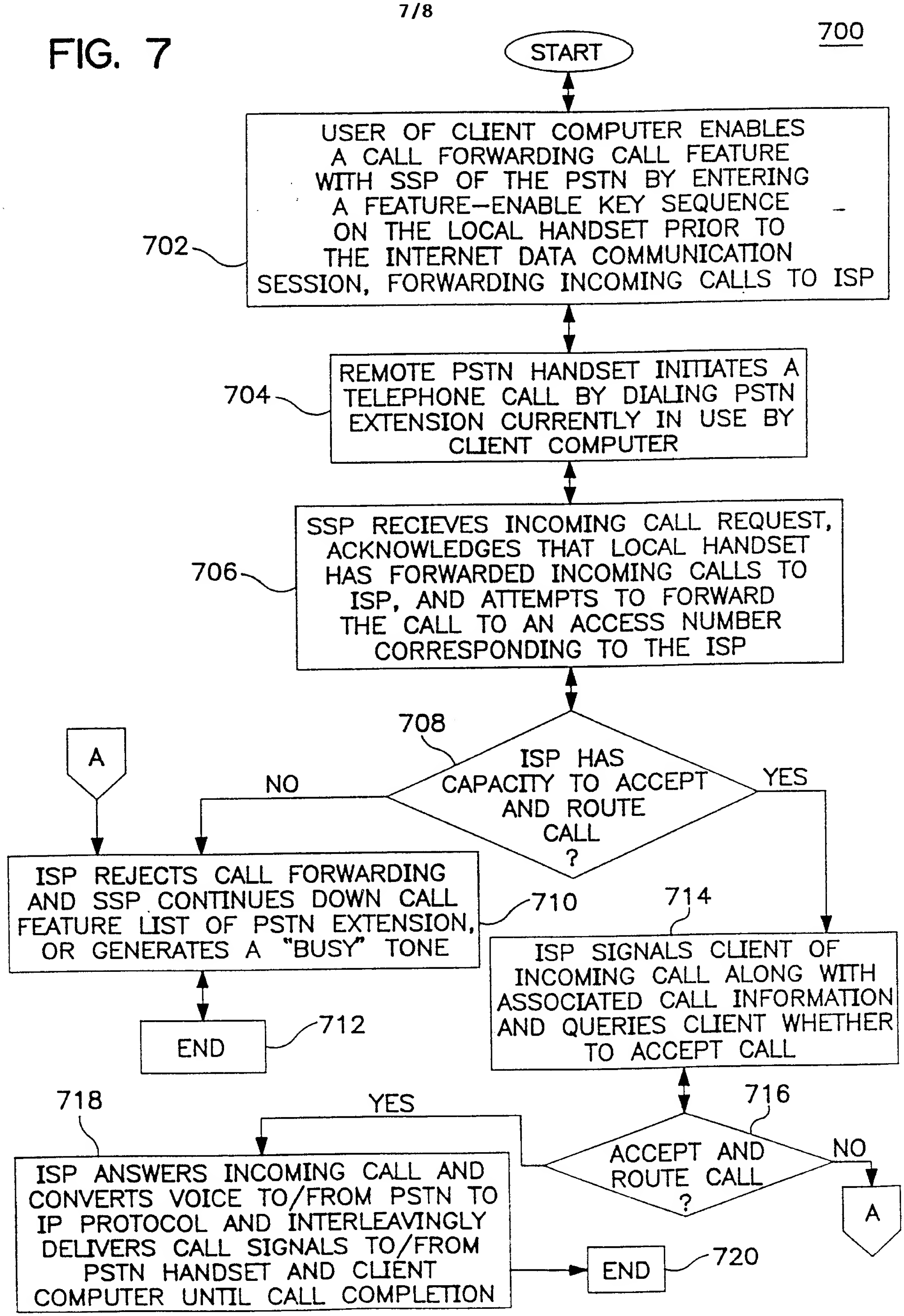

600
CPS
157
INTERNET
150
146
ISP
120
602
122
604
114
AP
112
126
STP
130
STP
134
SCP
HANDSET
140
108
110
SSP
136
SSP
170
106
104
CLIENT COMPUTER
102

FIG. 6

FIG. 7


700
START
USER OF CLIENT COMPUTER ENABLES A CALL FORWARDING CALL FEATURE WITH SSP OF THE PSTN BY ENTERING A FEATURE-ENABLE KEY SEQUENCE ON THE LOCAL HANDSET PRIOR TO THE INTERNET DATA COMMUNICATION SESSION, FORWARDING INCOMING CALLS TO ISP
702
REMOTE PSTN HANDSET INITIATES A TELEPHONE CALL BY DIALING PSTN EXTENSION CURRENTLY IN USE BY CLIENT COMPUTER
704
SSP RECIEVES INCOMING CALL REQUEST, ACKNOWLEDGES THAT LOCAL HANDSET HAS FORWARDED INCOMING CALLS TO ISP, AND ATTEMPTS TO FORWARD THE CALL TO AN ACCESS NUMBER CORRESPONDING TO THE ISP
706
708
ISP HAS CAPACITY TO ACCEPT AND ROUTE CALL ?
NO
YES
A
ISP REJECTS CALL FORWARDING AND SSP CONTINUES DOWN CALL FEATURE LIST OF PSTN EXTENSION, OR GENERATES A "BUSY" TONE
710
END
712
714
ISP SIGNALS CLIENT OF INCOMING CALL ALONG WITH ASSOCIATED CALL INFORMATION AND QUERIES CLIENT WHETHER TO ACCEPT CALL
716
ACCEPT AND ROUTE CALL ?
YES
NO
A
718
ISP ANSWERS INCOMING CALL AND CONVERTS VOICE TO/FROM PSTN TO IP PROTOCOL AND INTERLEAVINGLY DELIVERS CALL SIGNALS TO/FROM PSTN HANDSET AND CLIENT COMPUTER UNTIL CALL COMPLETION
END
720

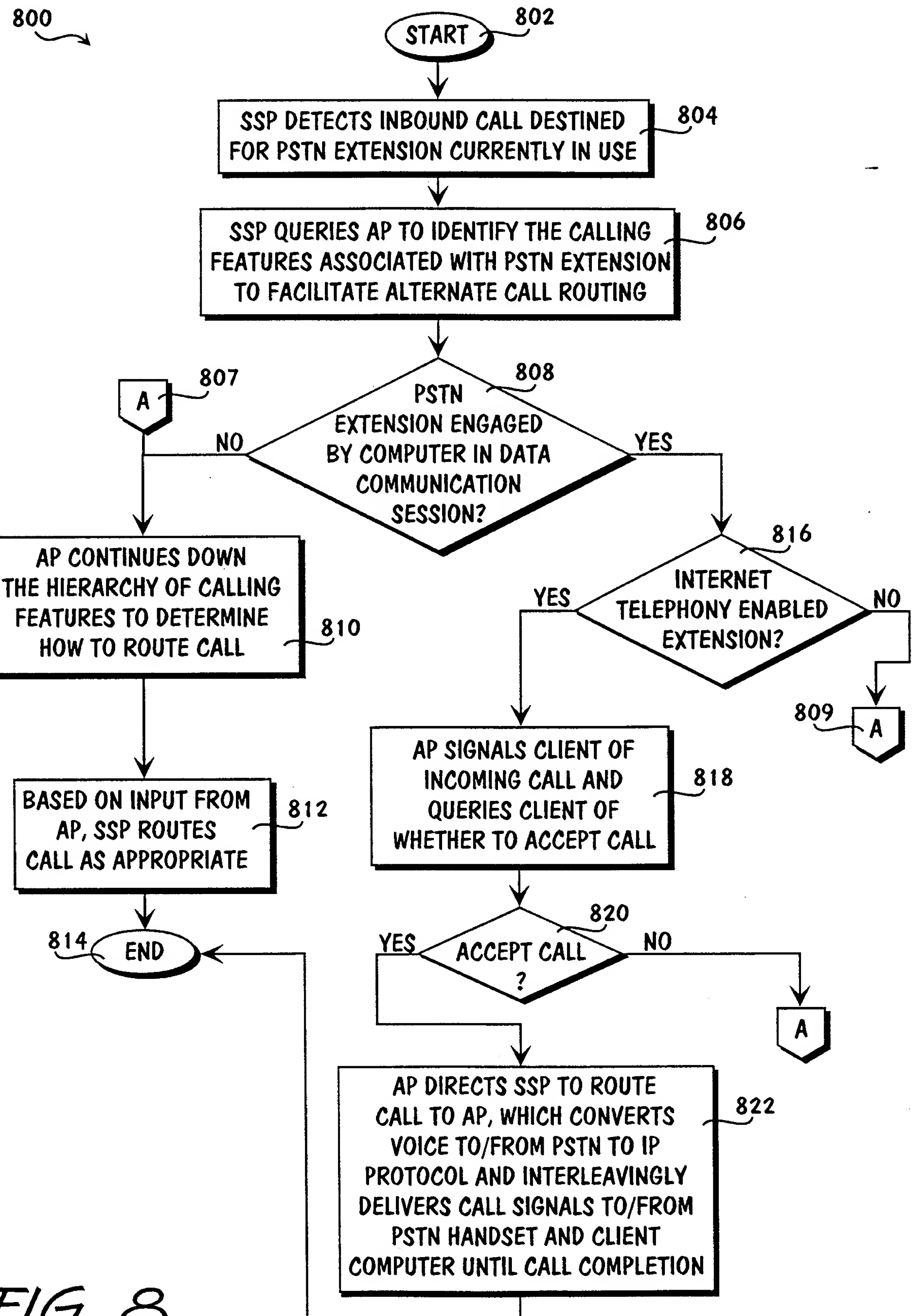

800
START
802
SSP DETECTS INBOUND CALL DESTINED FOR PSTN EXTENSION CURRENTLY IN USE
804
SSP QUERIES AP TO IDENTIFY THE CALLING FEATURES ASSOCIATED WITH PSTN EXTENSION TO FACILITATE ALTERNATE CALL ROUTING
806
A
807
PSTN EXTENSION ENGAGED BY COMPUTER IN DATA COMMUNICATION SESSION?
808
NO
YES
AP CONTINUES DOWN THE HIERARCHY OF CALLING FEATURES TO DETERMINE HOW TO ROUTE CALL
810
INTERNET TELEPHONY ENABLED EXTENSION?
816
YES
NO
809
A
AP SIGNALS CLIENT OF INCOMING CALL AND QUERIES CLIENT OF WHETHER TO ACCEPT CALL
818
BASED ON INPUT FROM AP, SSP ROUTES CALL AS APPROPRIATE
812
814
END
ACCEPT CALL ?
820
YES
NO
A
AP DIRECTS SSP TO ROUTE CALL TO AP, WHICH CONVERTS VOICE TO/FROM PSTN TO IP PROTOCOL AND INTERLEAVINGLY DELIVERS CALL SIGNALS TO/FROM PSTN HANDSET AND CLIENT COMPUTER UNTIL CALL COMPLETION
822

FIG. 8

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/21788

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04M 3/42; H04M 11/00
US CL :Please See Extra Sheet.
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 370/351, 352, 389; 379/67, 88, 89, 90.01, 93.01, 201, 209, 211, 215

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5,805,587 A (NORRIS et al) 08 September 1998, see entired document.	1-15
X	US 5,809,128 A (MCMULLIN) 15 SEPTEMBER 1998, See entired document.	1-15

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search

08 NOVEMBER 1999

Date of mailing of the international search report

14 DEC 1999

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/21788

A. CLASSIFICATION OF SUBJECT MATTER:
US CL :

370/351, 352, 389; 379/67, 88, 89, 90.01, 93.01, 201, 209, 211, 215